Section I. (Amendment of the Specification)

Please amend the specification as set out below.

Page 1, replace the paragraph at lines 1-8 with the following new paragraph:

The present invention relates in general to the ubiquity of the Internet Web Page for information and the ubiquity of the telephone for quality voice communications and in particular a method of converging these two ~~entity's~~ entities with a capability of connecting a Web Page user (Calling Party) and the Called Party (Web Page advertiser or single party subscriber) via a Managed Voice-over-Internet Protocol Network (MVoIP) that provides carrier-grade voice quality, and performance equal to the existing Public Switched Telephone Network (PSTN).

Page 5, replace the paragraph at lines 2-14 with the following new paragraph:

These prompts may take up to 35 seconds or more. After one has chosen to make a reservation and pressed (2), the call is placed in a queue and the next available, specialized agent will be given that call to answer. This queue timing is dependent on the number of agents available and how busy they are at the time the incoming call is received. Thus, the queue timing may range anywhere from five seconds to several minutes or longer. For purposes of the present explanation, a queue time of 15 seconds will be used. As identified in this example, the prompts may be listened to for 35 seconds and the queue time for waiting for an agent is 15 seconds. This represents a total of 50 seconds or more of billing time to ~~any~~ the business customer before the Calling Party starts actual conversation with an agent. During this time costly switch ports and voice circuit facilities are also being used prior to actual dialogue with an ~~advertisers~~ advertiser's agent (Called Party).

Pages 7-8, replace the paragraph beginning at page 7, line 7 and ending at page 8, line 3, with the following new paragraph:

Existing art "call me" capability will use two different methods of voice communications with the Web user. The first requires the implementation of first equipment (PC) that interfaces the Internet and second equipment (a telephone) that has direct connections to the PSTN. From that point on, i.e., when the telephone is used, the call is a typical PSTN call with all the problems of cost and inconvenience to the Calling Party. This implementation actually requires two PSTN calls to be made: a

device (server computer) in the PSTN places a call to the phone number of the Web Page user (Calling Party) over the PSTN and also places a call to the ~~Calling~~ Called Party. This doubles the communication cost to the business just to get the Web Page user to communicate person-to-person with the business. If the Web Page user (browser) has two telephones, he may use one of the phones to call the 800 toll-free number through the PSTN as described previously while the PC remains coupled to the Internet to receive IP data. If the Web Page user has only one phone line and is connected through it to the Internet, the user can either disconnect from the Internet and place the call on the 1-800 call through the normal PSTN, or can place the 1-800 call to the service Call Center 20 using the "call me" capability on the Web Page[[ ]]. The connection will be made through Internet connection 59 from the Internet network 54 to the local ISP access 60 in LSAP switch 22. The CPE equipment, in this case the Call Center ~~agents~~ agent's equipment 20, must be Voice-over-Internet Protocol compatible and enabled. The CPE equipment at the Calling Party CPE 12 must also be Voice-over-Internet Protocol compatible and enabled.

Page 8, replace the paragraph at lines 16-19 with the following new paragraph:

As can be seen in FIG. 2, in the prior art, the Internet 54 could also place a call on line 57 to an international LSAP using its ISP local access 44 which has coupled to it CPE 46 which may include a telephone coupled to a PC or ~~advertisers~~ advertiser's call agents connected to a Call Center.

Page 9, replace the paragraph at lines 6-9 with the following new paragraph:

The existing method of calling over the PSTN does not deliver an informed customer to the ~~advertisers~~ advertiser's CPE. Present art only allows a customer to be prescreened to the effect of the voice prompts made available to the Calling Party.

Pages 9-10, replace the paragraph beginning at page 9, line 12 and ending at page 10, line 18, with the following new paragraph:

Present state-of-the-art requires that the Web Page user have a PC capable of supporting compatible Voice-over-Internet Protocols such as H-323 standards. Not all Web Page users have microphones, speakers, and the software available on their PC to support this prior art, thus it is not ubiquitous. Most businesses do not provide multi-media ~~PC's~~ PCs to their employees. However, almost all employees have telephones and computers with ~~internet~~ Internet access. The present Internet has no

guaranteed delivery of service end-to-end because present Internet networks are designed to be data networks, not voice and data networks. Also, voice quality, as it presently exists on the Internet, is far too inferior and unpredictable for any type of business application, which requires communications between the Web Page user and the Web Page advertiser. Business customers expect the same quality of voice and reliability as is presently available over the PSTN. Further, the prior art system does not allow for total utilization of Internet Protocol (IP) technology such as queuing Web Page users in "cyber space" while waiting for an available advertiser agent to become available. Prior art implementation in some cases connects the Web Page user with the advertiser agent by placing calls through the PSTN, which provides the same problems as previously described in the prior art with calls on the PSTN. In addition, the prior art does not allow for coordination between the Web Page and CPE Call Center procedures. With the prior art there is clearly no ubiquity in service due to not all the CPE of Calling Parties having multimedia capability, i.e., microphones, speaker, Voice-over-IP software compatibility. In addition, prior art implementations allow access to advertiser agents through use of chat, e-mail, etc., which do not solve cultural issues such as the need to have personal voice contact when completing a transaction. Existing art is based on the concept of callback for voice communications. This is because the prior art is based on an implementation where the agent gets the Web user's information such as telephone number, from the chat ~~secession~~ session, the e-mail received, or the information received on a Web-enabled agent screen. The agent then places a call to the Calling Party, thus the "callback" scenario. This creates problems in coordinating the callback with the availability of the Calling Party. This method of implementation does not allow for instant buying by the Calling Party.

Pages 13, replace the paragraph at lines 1-20 with the following new paragraph:

The present invention enables the Calling Person (user), through Web Page navigation, to make selections of products and/or services advertised on the [[Inter- net]] Internet and to provide detailed information to the Internet system concerning such selections along with a customer profile. The invention provides for informed Web Page users to communicate with an agent by forwarding the data completed by the Web Page user (Calling Party) to the agent at the time of answer by the agent. This data can include such information as name, address, phone number, age, language preference, and other product-related information as requested through Web Page navigation custom-made for the particle customer business. This reduces agent work time, conversation time, and produces a satisfied and content customer (Calling Party). Web Page design and navigation can provide all the data needed by the agent to complete the transaction. This capability is not available over the existing PSTN. The novel system allows the call of a Calling Person to automatically navigate through any "prompts", identify the

information required to purchase the product/service such as flight information, flight number, number of people traveling, preferred seating, international or domestic flight preference, Advantage card numbers, billing information, color, size, and the like, and thus, delivers an informed customer to the agent of the Web Page advertiser (Called Party).

Pages 13-14, replace the paragraph beginning at page 13, line 21 and ending at page 14, line 2, with the following new paragraph:

The invention also allows "call me" capability from any number associated with a Web Page, whether it be to a Call Center or to a single telephone. For example, a business may offer a Web Page that contains a "call me" capability to their Call Center for product purchases or service while at the same time ~~have~~ has phone numbers associated with a receptionist ~~of~~ or an individual.

Pages 15-16, replace the paragraph beginning at page 15, line 4 and ending at page 16, line 2, with the following new paragraph:

This also allows a Web Page user to click on a Web Page equipped with "call me" capability and a call will be placed to the ~~advertisers~~ advertiser's agent without the use of 800 numbers. The call will cost significantly less due to the use of a MVoIP network connecting the ~~LSAP's~~ LSAPs internationally with the LSAP of the advertiser (Called Party).

Page 20, replace the paragraph at lines 11-13 with the following new paragraph:

FIG. 3 is a schematic representation of the Public Switching Telephone Network and the Internet system modified by a Managed Voice-over-Internet Protocol ~~network (MVIOP)~~ (MVoIP) network of the present invention; and

Page 21, replace the paragraph at lines 3-25 with the following new paragraph:

The novel system 62 of FIG. 3 includes not only the Public Switching Telephone Network (PSTN) 30 and its associated operating elements, but also the Internet 54 and its associated elements coupled to a Managed Voice-over-Internet Protocol network (MVoIP) 64. The MVoIP network 64 includes a first gateway 66 that couples both voice and IP data between the first LSAP 14 and second LSAP 22 coupled to the Call Center 20. It also includes a second gateway 68 that couples both voice and

IP data on line 80 to second LSAP 22 coupled to Call Center 20. International gateway 102 is also coupled to MVoIP network on line 82. The international gateway 102 is coupled to international LSAP 44 on line 103. International CPE 104 is coupled to LSAP 44 on line 105. ~~LSAP's~~ LSAPs 14[[ ]], 22 and 44 have access to the Internet on connections[[,]] 55, 59 and 57. Gateways 66, 68, and 102 are well known in the art and **are defined herein** as systems that provide translation of protocols for call setup and release, conversion of media formats between different networks, and transferring of information between networks connected by the gateway such as LSAP 14[[ ]], LSAP 22, and LSAP 44. The gateways 66 and 68 used in the present invention operate with the well-known Voice-over-IP standards that are cornerstone technology for the transmission of real-time audio, video, and data communications over packet-based networks. VoIP **is defined as a standard that** specifies the components, protocols, and procedures providing multimedia communication over packet-based networks. Because these gateways meet the VoIP standards, no further discussion will be provided regarding them.

Page 24, replace the paragraph at lines 7-12 with the following new paragraph:

It will be seen, then, that the MVoIP network provides reliable carrier-grade voice quality service over connections established between the parties using the Internet. The MVoIP network provides VoIP gateway functions between the ~~LSAP's~~ LSAPs and the IP networks. As stated earlier, functions of the gateways in the MVoIP including translating IP and PSTN addresses, covering network interface bridge protocol and the like.

Page 26, replace the paragraph at lines 6-20 with the following new paragraph:

A flow chart illustrating the relationship of the system elements and the novel steps of the present invention is shown in FIG. 4A and FIG. 4B. This flow chart shows the typical Internet connection to browse a Web Page to view products/services, then using a Web server "call me" button to establish a VoIP connection over a reliable voice quality MVoIP network to the CPE [[ ]] Call Center (Called Party) or a single phone, then connect to the original single or multi-line telephone Internet user/Calling Person. It can be seen at the top of FIG. 4A that the PSTN is used in the normal fashion to connect a Calling Party LSAP to a Called Party LSAP. Further, it can be seen at the top of FIG. 4A that the Internet can be used to connect a Calling Party to a Called Party through an Internet Service Provider (ISP). This connection for voice communication is very unreliable and very poor quality. This service is available "for free" and is not being used for business application ~~were~~ where quality and reliable service is required. Further, this service is not ubiquitous to all Web Page users.

Page 28, replace the paragraph at lines 3-9 with the following new paragraph:

If no agents are available, the call is placed in a queue at step 16 to await the ~~--------------------~~ next available agent. When an agent becomes available, the off-hook notice is given at step 17 and a communication link is established between the agent and the CCS at step 18. At step 19, the CCS ~~initiate~~ <u>initiates</u> a "callback" to the Calling Person and a "ring" connection is made to the Calling Party through the LSAP and the CPE of the Calling Person at step 20.